(19)日本国特許庁（ＪＰ）　(12) 公開特許公報（Ａ）　(11)特許出願公開番号

特開平8－110800

(43)公開日　平成８年(1996)４月30日

(51)Int.Cl.6		識別記号	庁内整理番号	ＦＩ	技術表示箇所
Ｇ１０Ｌ	9/18	Ｅ			
	9/14	Ｊ			

審査請求　未請求　請求項の数５　ＯＬ　（全 ９ 頁）

(21)出願番号　特願平6－246504

(22)出願日　平成６年(1994)10月12日

(71)出願人　000005223
富士通株式会社
神奈川県川崎市中原区上小田中1015番地

(72)発明者　向　厚幸
福岡県福岡市博多区博多駅前三丁目22番８号　富士通九州ディジタル・テクノロジ株式会社内

(74)代理人　弁理士　古谷　史旺　（外１名）

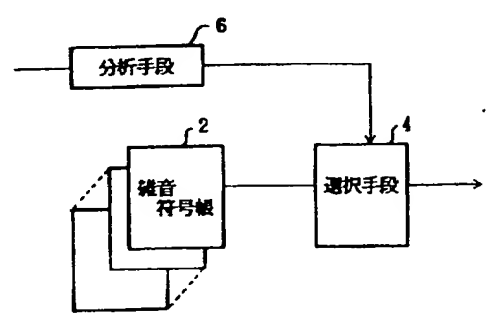
(54)【発明の名称】　Ａ－ｂ－Ｓ法による高能率音声符号化方式

(57)【要約】

【目的】　本発明は、Ａ－ｂ－Ｓ法による高能率音声符号化方式に関し、背景雑音がある状態における通話においても通話に違和感を生じさせることなく通話品質の良い状態で通話することを目的とする。

【構成】　Ａ－ｂ－Ｓ法による高能率音声符号化方式において、入力音声信号の周波数特性に応じた雑音の符号ベクトルを格納する予め決められた数の雑音符号帳と、各雑音符号帳の出力に接続された選択手段と、入力音声の周波数特性を分析して分析された周波数特性に応じて選択手段の切替えを生ぜしめる分析手段を設け、選択手段の出力を入力音声の符号化に用いる雑音符号帳として使用することを特徴とする。

請求項１、請求項２及び請求項５記載の発明の原理ブロック図



(11) Japanese Patent Laid-Open No. 110800/1996

(43) Laid-Open Date: April 30, 1996

(21) Application No. 246504/1994

(22) Application Date: October 12, 1994

(71) Applicant: Fujitsu Kabushiki Kaisha

(72) Inventor: Atsuyuki(?) MUKAI

Request for Examination: Not made

(54) [Title of the Invention] HIGH-EFFICIENCY VOICE ENCODING SYSTEM USING A-b-S METHOD

(57) [Abstract]

[Object] The present invention relates to high-efficiency voice encoding systems using A-b-S methods, and its object is to enable establishment of a telephone call under the condition that online speech quality is excellent without occurring any sense of discomfort even in cases where the telephone call is done in the presence of background noises.

[Structure] A high-efficiency voice encoding system using A-b-S methodology is disclosed which is characterized by providing a predetermined number of noise code books each storing therein more than one noise code vector in accordance with the frequency characteristics of an input audio or voice signal, selector means connected to an output of each noise code book, and analysis means for analyzing the frequency

characteristics of input voice to thereby permit generation of changeover of the selector means in accordance with the frequency characteristics thus analyzed, wherein an output of the selector means is used as a noise code book which is used for encoding of input voice.

[Claims]

[Claim 1] A high-efficiency voice encoding system using an A-b-S method, characterized by providing:

a predetermined number of noise code books each storing therein a noise code vector in accordance with the frequency characteristics of an input voice signal;

selector means connected to an output of each noise code book; and

analysis means for analyzing the frequency characteristics of input voice and for permitting creation of changeover of the selector means in accordance with the frequency characteristics thus analyzed, wherein

an output of said selector means is used as a noise code book being used for encoding of the input voice.

[Claim 2] The high-efficiency voice encoding system using an A-b-S method as recited in claim 1, characterized in that when a difference between a maximal value and a minimal value of a linear prediction coding coefficient in the A-b-S method falls within a predetermined threshold value range,

said analysis means causes selection of a noise code book corresponding to the threshold value range in said selector means.

[Claim 3] An A-b-S method-based high-efficiency voice encoding system having a background noise removal device, characterized by providing:

an inverse filter responsive to receipt of a prediction coefficient from a noise prediction filter of said background noise removal device for outputting a predicted noise signal;

a power integrator for integration of power of the noise signal as output from the inverse filter; and

a subtractor for outputting a value of the noise signal from said inverse filter as subtracted by a power integration value from said power integrator, wherein

said subtracted value is used as the noise code vector of a noise code book.

[Claim 4] An A-b-S method-based high-efficiency voice encoding system having a background noise removal device and a linear prediction coding analysis unit along with a pitch gain adjuster and a noise gain adjuster, characterized by providing:

an inverse filter responsive to receipt of a prediction coefficient from a noise prediction filter of said background noise removal device for outputting a predicted noise signal;

a power integrator for integration of power of the noise signal as output from the inverse filter; and

a coefficient calculation unit for using a power value from said linear prediction coding analysis unit and a power integration value from said power integrator to thereby output a noise gain coefficient to said noise gain adjuster and a pitch gain coefficient to said pitch gain adjuster.

[Claim 5] The A-b-S method-based high-efficiency voice encoding system as recited in any one of the preceding claims 1 to 3, characterized by providing:

an inverse filter responsive to receipt of a prediction coefficient from the noise prediction filter of said background noise removal device for outputting a predicted noise signal;

a power integrator for integration of power of the noise signal as output from the inverse filter; and

a coefficient calculation unit for using a power value from said linear prediction coding analysis unit and a power integration value from said power integrator to thereby output a noise gain coefficient to said noise gain adjuster and a pitch gain coefficient to said pitch gain adjuster, wherein

the code vector of a pitch of an adaptive code book is adjusted by the pitch gain adjuster to an extent that corresponds to a pitch gain coefficient as output from said coefficient calculation unit whereas the code vector of a noise of said noise code book is adjusted by the noise gain adjuster to an extent that corresponds to the noise gain coefficient as output from said coefficient calculation unit.

[Detailed Description of the Invention]

[0001]

[Field of Industrial Application] The present invention relates to A-b-S method-based high-efficiency speech/voice encoding systems using a noise code book in accordance with the frequency characteristics of an input voice signal for voice signal encoding.

[0002] In mobile telephone handsets, an audio or voice signal used for a telephone call thereof is not directly transmitted and received as a call signal with no modifications applied thereto but is sent and received after execution of encoding processing. Such coding is especially based on high-efficiency voice encoding systems using so-called "A-b-S" methodology. Although the coding systems of this type are typically designed to perform encoding of voice signals in a way as will be described later, the encoding still has several points to further improve, depending upon environments for performing telephone calls.

[0003]

[Prior Art] A high-efficiency voice encoding system employing the above-noted A-b-S method has an arrangement as shown in Fig. 7, which encodes an audio/voice signal in a way which follows. An input voice signal is such that the frequency characteristics of the input voice signal is computed by an LPC analysis unit 70 with a predetermined length of frame time

period as a unit to thereby obtain an LPC coefficient and a filter coefficient of a synthetic filter 72 while at the same time obtaining a power component which is then used to determine the gain of code vectors that will be output from an adaptive code book 74 and a noise code book 76. This gain is given as a ratio of the adaptive code book to the noise code book with respect to a power integration value of the input voice signal. It should be noted here that the term "LPC" is an abbreviation of Linear Predicative Coding.

[0004] A respective one of the code vectors as stored in the adaptive code book 74 and noise code book 76 is read in units of said frame time periods. This readout will be done in a way as will be described later. The code vectors thus read are adjusted in gain by a gain coefficient which is determinable depending on said power value obtained at gain adjusters 78, 80; thereafter, the both code vectors are added together at an adder 90 and are then supplied to a synthetic filter 72. The code vectors thus input to the synthetic filter 72 are combined or synthesized together at the synthetic filter 72 that receives a filter coefficient from the LPC analysis unit 70 and are then output as a synthesized voice signal.

[0005] The synthetic voice signal is supplied to a subtractor 94; then, a difference of it from the input voice signal will be output from the subtractor 94. This difference is supplied to an error minimization processing unit 96, wherein code

vectors for minimization of such difference are read out of the adaptive code book 74 and the noise code book 76 and then used for the aforementioned gain adjustment. For instance, in the event that the adaptive code book 74 and noise code book 76 are the ones that have 1,024 code vectors respectively, certain code vectors out of 1024 × 1024 combinations which minimize said difference are read out of the adaptive code book 74 and noise code book 76 in units of frame time periods as the code vectors to be used for the A-b-S method-based high-efficiency voice encoding.

[0006] In units of respective frame time periods in the aforesaid processing, the power component signal, address of each code book, gain coefficient to be multiplied to the code vectors as output from the adaptive code book and the noise code book, and LPC coefficient will be sent toward a receipt destination as coding data in the A-b-S method-based high-efficiency voice encoding system.

[0007]

[Problem to be Solved by the Invention] Although the code vector of the adaptive code book in the above-noted prior art high-efficiency voice encoding system is to be adaptively updated in accordance with an input voice signal, the code vector being stored in the noise code book is the one that is fixed to pre-simulated white noises or the like. The code vector of the noise code book is fixed in this way because it

specifies as the characteristics owned by a class of speech sounds, in particular "consonants," of voice signals.

[0008] With the adaptive code book thus arranged in the way stated above, it will no longer be adapted in real telephone call environments. This can be said because noises occurring in real telephone call environments are random noises containing background noises, which are far different from fixed noises as represented by the fixed code vector being stored in the above-noted noise code book.

[0009] Accordingly, in real telephone call environments, any audio/voice signals caught by recipients become unnatural hard-to-hear voice signals, which in turn retards or impedes telephone call activities. The present invention was made in view of the above technical problem and its primary object is to provide a high-efficiency voice encoding system using the A-b-S methodology capable of offering an ability to achieve a telephone call of good speech quality without creating any sense of discomfort even in telephone call events with the presence of background noises.

[0010]

[Means for Solving the Problem] Fig. 1 shows a principal block diagram of the inventions as recited in the claim 1, claim 2, and claim 5. Fig. 2 shows a principle block diagram of the inventions as recited in claim 3 and claim 5. Fig. 3 is a

principle block diagram of the inventions as recited in claim 4 and claim 5.

[0011] The invention as recited in claim 1 is such that as shown in Fig. 1, in a high-efficiency voice encoding system using an A-b-S method, the system is characterized by providing a predetermined number of noise code books 2 each storing therein more than one noise code vector in accordance with the frequency characteristics of an input audio/voice signal, selector means 4 connected to an output of each noise code book, and analysis means 6 for analyzing the frequency characteristics of input voice and for permitting creation of changeover of the selector means in accordance with the frequency characteristics thus analyzed, wherein an output of said selector means is used as a noise code book being used for coding of the input voice.

[0012] The invention as recited in claim 2 is such that as shown in Fig. 1, in the A-b-S method-based high-efficiency voice encoding system, the system is characterized in that when a difference between the maximum value and minimum value of a linear prediction coding coefficient in the A-b-S method falls within a predetermined threshold value range, said analysis means 6 causes selection of a noise code book corresponding to the threshold value range in said selector means.

[0013] The invention as recited in claim 3 is such that as shown in Fig. 2, in an A-b-S method-based high-efficiency voice encoding system having a background noise removal device 8,

this system is characterized by providing an inverse filter 10 which is responsive to receipt of a prediction coefficient from a noise prediction filter of said background noise removal device 8 for outputting a predicted noise signal, a power integrator 12 for integration of power of the noise signal as output from the inverse filter 10, and a subtractor 14 for outputting a value of the noise signal from said inverse filter 10 as subtracted by a power integration value from said power integrator 12, wherein said subtracted value is used as the noise code vector of a noise code book.

[0014] The invention as recited in claim 4 is such that as shown in Fig. 3, in an A-b-S method-based high-efficiency voice encoding system having a background noise removal device 8 and a linear prediction coding analysis unit 16 along with a pitch gain adjuster 20 and a noise gain adjuster 21, the system is characterized by providing an inverse filter 10 which is responsive to receipt of a prediction coefficient from a noise prediction filter of said background noise removal device 8 for outputting a predicted noise signal, a power integrator 12 for integration of power of the noise signal as output from the inverse filter 10, and a coefficient calculation unit 18 for using a power value from said linear prediction coding analysis unit 16 and a power integration value from said power integrator 12 to thereby output a noise gain coefficient to said noise gain

adjuster 21 and a pitch gain coefficient to said pitch gain adjuster 20.

[0015] The invention as recited in claim 5 is such that as shown in Fig. 1, Fig. 2 and Fig. 3, in the A-b-S method-based high-efficiency voice encoding system as recited in any one of the preceding claims 1 to 3, it is characterized by providing an inverse filter 10 which is responsive to receipt of a prediction coefficient from the noise prediction filter of said background noise removal device 8 for outputting a predicted noise signal, a power integrator 12 for integration of power of the noise signal as output from the inverse filter 10, and a coefficient calculation unit 18 for using a power value from said linear prediction coding analysis unit 16 and a power integration value from said power integrator 12 to thereby output a noise gain coefficient to said noise gain adjuster 21 and a pitch gain coefficient to said pitch gain adjuster 20, wherein the code vector of a pitch of an adaptive code book is adjusted by the pitch gain adjuster 20 to an extent that corresponds to a pitch gain coefficient as output from said coefficient calculation unit 18 whereas the code vector of a noise of said noise code book is adjusted by the noise gain adjuster 21 to an extent that corresponds to the noise gain coefficient as output from said coefficient calculation unit 18.

[0016]

[Operation] The invention as recited in claim 1 is such that in the high-efficiency voice encoding process using the A-b-S method of an input voice signal, it analyzes by the analysis means the frequency characteristics of such input voice signal. It selects by the selector means a noise code book as determinable depending upon said analysis result from among the plurality of noise code books as prepared in advance.

[0017] With this selection, it is possible to use the noise code book which is adapted to the frequency characteristics of the input voice signal for effectuation of the A-b-S method-based high-efficiency voice encoding; thus, even in cases where a telephone call environment is out of the ideal telephone call environment, it is possible to hear and catch the speech or voice of the transmitter side without occurrence of any uncomfortable feelings as to voices being heard and caught on the receiver side. Thus it is possible to improve the online speech quality.

[0018] The invention as recited in claim 2 is the one in which the analysis due to the analysis means of the invention as recited in claim 1 is done in such a way as to determine which one is selected from among the plurality of noise code books in accordance with a difference between the maximum value and minimum value of the linear prediction coding coefficient in the A-b-S method.

[0019] The invention as recited in claim 3 is the one that calculates the noise vector of a noise code book to be used for high-efficiency voice encoding by the A-b-S method from more than one prediction coefficient as obtainable from the background noise removal device and then uses such noise vector for high-efficiency voice encoding by the A-b-S method. Even in cases where a present telephone call environment is out of the ideal telephone call environment, it is possible to hear and catch any voices on the transmitter side without occurring any sense of discomfort as to voices being heard on the receiver side. Thus it is possible to improve the online voice quality.

[0020] The invention as recited in claim 4 is the one that introduces the pitch gain coefficient to be used at the pitch gain adjuster and the noise gain coefficient being used at the noise gain adjuster from a prediction coefficient which is obtainable from the background noise removal device. These pitch gain coefficient and noise gain coefficient are used to adjust the gain of a respective one of coding vectors as output from the adaptive code book and noise code book for execution of high-efficiency voice encoding by the A-b-S method; thus, even in cases where a present telephone call environment is out of the ideal telephone call environment, it is possible to hear and catch any voices on the transmitter side without occurring any uncomfortable feelings as to voices being heard on the

receiver side. Thus it is possible to improve the online voice quality.

[0021] The invention as recited in claim 5 is the one that adjusts respective code vectors being output from the noise code book and adaptive code book which are used in the invention as set forth in claim 1, claim 2 or claim 3 by the pitch gain coefficient and noise gain coefficient that are obtainable by the invention as recited in claim 4. In this case also, even in cases where a present telephone call environment is out of the ideal telephone call environment, it is possible to hear and catch any voices on the transmitter side without occurring uncomfortable feelings as to voices being heard on the receiver side. Thus it is possible to improve the online voice quality.

[0022]

[Embodiments] Fig. 4 shows one embodiment of the inventions as recited in claim 1, claim 2 and claim 5. A characteristic part of this embodiment lies in that during high-efficiency voice encoding based on an A-b-S method as shown in Fig. 7, a predetermined number of noise code books are prepared in advance, wherein LPC coefficient is used in a way as will be discussed below to thereby select a single noise code book from among said plurality of noise code books. The plurality of noise code books are arranged so that these are provided in accordance with the frequency characteristics of an input audio or voice signal. An example of it with provision of a plurality

of noise code books in accordance with the frequency characteristics of input voice signal will be indicated below.

[0023] More specifically, the noise code books are constituted from noise code books 71_1, 71_2,..., 76_N of certain noises such as a white noise (noise 1), noise obtained through trailing by simulation (noise 2), pink noise (noise 3) and the like. Reference numeral 23 designates a selector, and numeral 22 denotes a coefficient difference comparator unit which is operable to select the selector 23. The coefficient difference comparator unit 22 is supplied with an LPC coefficient from an LPC analysis unit 70, wherein the unit 22 determines whether a difference between an n-dimensional LPC coefficient whose value indicates a maximal value and (n+i)-dimensional LPC coefficient indicative of a minimal value falls within a predetermined difference threshold value range and then outputs a select signal for selection of a noise gain adjuster with said difference falling within the difference threshold value range.

[0024] One example is that as shown in Table 1 below, when the difference threshold value measures 0 to 2 dB, select the noise code book of white noise; if the difference threshold value is 3-7dB then select a noise code book corresponding to the noise as obtained through training by simulation; if the difference threshold value is 7-10dB then select a noise code book corresponding to pink noise.

[0025]

[Table 1]

Differential Threshold Value	Noise Code Book
0 – 2 dB	Noise 1 (White)
3 – 7 dB	Noise 2
⋮	⋮
7 – 10 dB	Noise 3 (Pink)

[0026] A respective one of the code vectors of a presently selected noise code book is written into the noise code book 25 for later use in high-efficiency voice encoding system based on the A-b-S method. A way of using it is similar to that in the prior art.

[0027] It should be noted that a way of using the adaptive code book for the A-b-S method-based high-efficiency voice encoding is substantially the same as that of the prior art A-b-S method-based high-efficiency voice encoding. And, in the A-b-S method-based high-efficiency voice encoding of this embodiment also, power signal components, addresses of the adaptive code book and noise code book, gain coefficient and LPC coefficient will be transmitted as coded data in a similar way to that of the prior art.

[0028] In Fig. 4, the noise code books 76_1, 76_2,..., 76_n correspond to the noise code books 2 of Fig. 1; the selector 23 corresponds to the selector means 4 of Fig. 1. The LPC

analysis unit 70 and coefficient difference comparator unit 22 correspond to the analysis means 6 of Fig. 1.

[0029] In this way, in the A-b-S method-based high-efficiency voice encoding system, since selective use of the noise code books makes it possible to use an appropriate noise code book being matched with the frequency characteristics of an input voice signal for coding thereof, mere transmission of a voice signal as coded by the above-stated A-b-S method-based high-efficiency voice encoding system toward the receiver side makes it possible to hear and catch voices on the transmitter side without occurrence of uncomfortable feelings as to voices being heard on the receiver side even in cases where a present environment is out of the ideal telephone call environment (telephone call environment with white noises).

[0030] Fig. 5 depicts an embodiment of the inventions as recited in claim 3 and claim 5. A characteristic part of this embodiment lies in that in the A-b-S method-based high-efficiency voice encoding system shown in Fig. 7, a noise code book which was generated from a prediction coefficient being used in a background noise removal device under development is used as the noise code book in A-b-S method-based high-efficiency voice encoding procedure.

[0031] In Fig. 5, numeral 30 designates an inverse filter which receives a prediction coefficient to be used in a not-illustrated background noise removal device 100 (not shown),

which filter has its input to which "0" is being input. 32 is a normalizer device 32. A normalization signal as output from this normalizer device 32 makes up the noise code book(s). The normalizer device 32 consists essentially of a power integrator 34 and a subtractor 36. Although Fig. 5 indicates only the characteristic part of the embodiment of the inventions as recited in claim 3 and claim 5, the remaining constituent elements are the same as the constituent elements that have been explained in conjunction with Fig. 7 so that the same reference numerals are added to the same constituent elements with any explanations thereof eliminated herein.

[0032] In Fig. 5, the background noise removal device 100 corresponds to the background noise removal device 8 of Fig. 2 whereas the inverse filter 30 corresponds to the inverse filter 10 of Fig. 2. The power integrator 34 corresponds to the power integrator 12 of Fig. 2; the subtractor 36 corresponds to the subtractor 14 of Fig. 2.

[0033] An explanation will be given of an operation of the high-efficiency voice encoding apparatus shown in Fig. 5 below. As this high-efficiency voice encoding apparatus is the same as the A-b-S method-based high-efficiency voice encoding of Fig. 4 in operation excluding outputting of a noise code vector from the noise code book, its detailed explanation will be omitted herein and an explanation will be given of certain part relating to the noise code book only.

[0034] The background noise removal device 100 shown herein is made up from a noise prediction filter and a subtractor which is operable to subtract a predicted noise as output from the noise prediction filter from an input audio or voice signal, and is the one that outputs a voice signal from the subtractor thereof. A noise prediction coefficient for use in such noise prediction filter is supplied to the inverse filter 30. A noise signal is output from this inverse filter 30 in units of frame periods. Let the noise signal component y(i) be integrated by the power integrator 34 in units of frame periods to thereby output a power integration value $\Sigma y(i)^2$, where "i" is the length of a vector. A prespecified number—for example, 1024—of noise code vectors obtainable by subtraction (normalization) between the noise signal component $y(i)^2$ and power integration value $\Sigma y(i)^2$ at the subtractor 36 will be written into the noise code book 36 in units of frame time periods at frame-period corresponding storage locations thereof: after having written a specified number, updating will be done sequentially.

[0035] It must be noted that a way of using the adaptive code book for the A-b-S method-based high-efficiency voice encoding is substantially the same as that of the prior art A-b-S method-based high-efficiency voice encoding. And, in the A-b-S method-based high-efficiency voice encoding of this embodiment also, power signal components, addresses of the adaptive code book and noise code book, gain coefficient and LPC coefficient

will be transmitted as coded data in a similar way to that of the prior art.

[0036] In the way stated above, in the A-b-S method-based high-efficiency voice encoding system, since it is possible to use the noise code book as generated in the way discussed above for the coding thereof, it becomes possible to obtain an appropriate noise code book adapted for a telephone call environment as the noise code book to be used in the above-noted A-b-S method-based high-efficiency voice encoding system; thus, mere transmission of a coded voice signal toward the receiver side makes it possible to hear and catch voices on the transmitter side without creation of any uncomfortable feelings as to voice being heard on the receiver side even in the event that a present environment is out of the ideal telephone call environment (telephone call environment with white noises).

[0037] Fig. 6 shows one embodiment of the inventions as recited in claim 4 and claim 5. A characteristic part of this embodiment is that in the A-b-S method-based high-efficiency voice encoding system shown in Fig. 7, a pitch gain coefficient and noise gain coefficient as generated from the prediction coefficient to be used in a background noise removal device under development and the power value being output from the LPC analysis unit 70 are used as the pitch gain coefficient and

noise gain coefficient in the A-b-S method-based high-efficiency voice encoding procedure.

[0038] In Fig. 6, numeral 40 denotes an inverse filter to which a prediction coefficient being used in a prediction noise filter of the background noise removal device 100, not depicted, is supplied, wherein a noise signal is output from this inverse filter 40 in units of sampling (frame) periods. Such noise signal $y(i)^2$ is supplied to the power integrator 42. A power integration value $\Sigma y(i)^2$ being output from the power integrator 42 is supplied to a coefficient calculation unit 44 to be supplied to gain adjusters 78, 80, where "i" of the power integration value $\Sigma y(i)^2$ is a vector length. Supplied to this coefficient calculator unit 44 is a power value $\Sigma P(i)^2$ from the LPC analysis unit 70, where *i* of the power value $\Sigma P(i)^2$ is a vector length. The coefficient calculator unit 44 calculates in units of frame time periods a gain coefficient Ng(v) for execution of gain adjustment of a noise code vector as output from the noise code book after division of the power integration value $\Sigma y(i)^2$ by power value $\Sigma P(i)^2$ and then calculates in units of frame time periods a gain coefficient Pg(v)=1-Ng(v) for execution of gain adjustment of a pitch code vector(s) as output from the adaptive code book. In place of these arithmetic processings, it may be arranged so that previously obtained data through simulation is prepared in advance in a table format and this table is indexed to obtain the gain coefficients Pg(v)

and Ng(v). Here, v of the gain coefficients Pg(v) and Ng(v) is a vector number. And, the gain coefficient Pg(v) is supplied to the gain adjuster 78 whereas the gain coefficient Ng(v) is supplied to the gain adjuster 80. The code vector as gain-adjusted at the gain adjuster 78 in units of frame time periods and the code vector as gain-adjusted at the gain adjuster 80 in units of frame time periods are supplied to the adder 90 in a way similar to that as has been explained with respect to Fig. 7. Its following operation is the same as that explained in Fig. 7 so that any detailed explanation is omitted herein. Note however that at the error minimization processing unit 96, the gain coefficients Pg(v) and Ng(v) obtainable in the way stated supra are handled as parameters for minimization of said difference in a similar way to that of the pitch code vector and noise code vector.

[0039] Optionally the noise code book may be a fixed noise code book as in the prior art; it may alternatively be the noise code book with the arrangement shown in Fig. 4; or still alternatively, it may be the noise code book with the arrangement shown in Fig. 5.

[0040] And, coding data of the A-b-S method-based high-efficiency voice encoding system may also include said gain coefficients Pg(v) and Ng(v) in addition to the power component, addresses of the adaptive code book and noise code book and LPC coefficient.

[0041] Additionally a way of using the adaptive code book in this embodiment for the A-b-S method-based high-efficiency voice encoding is substantially the same as that of the prior art A-b-S method-based high-efficiency voice encoding. In Fig. 6, the background noise removal device 100 corresponds to the background noise removal device 8 of Fig. 3; the inverse filter 40 corresponds to the inverse filter 10 of Fig. 3. The power integrator 42 corresponds to the power integrator 12 of Fig. 3. The LPC analysis unit 70 corresponds to the linear prediction coding analysis unit 16 of Fig. 3; the coefficient calculator unit 44 corresponds to the coefficient calculator unit 18 of Fig. 3. The pitch gain adjuster 78 corresponds to the pick gain adjuster 20 of Fig. 3; the noise gain adjuster 80 corresponds to the noise gain adjuster 21 of Fig. 3.

[0042] In this embodiment, in the A-b-S method-based high-efficiency voice encoding, since the gain coefficient Pg(v) that was calculated in the way discussed above is used to perform the gain adjustment of pitch code vector while using the gain coefficient Ng(v) to perform the gain adjustment of noise code vector, the pitch code vector and noise code vector being used in the above-stated A-b-S method-based high-efficiency voice encoding system are such that it is possible to obtain appropriate pitch code vector and noise code vector adapted for a telephone call environment; thus, mere transmission of a coded audio/voice signal toward the receiver

side makes it possible to hear and catch voices on the transmitter side without creation of any uncomfortable feelings as to the voice being heard on the receiver side even in the event that a present environment is out of the ideal telephone call environment (telephone call environment with white noises).

[0043]

[Effects of the Invention] It has been set forth that in accordance with the present invention, since any noise code book used is not a fixed one but a noise code book adapted to telephone call environments, it is possible to hear and catch any voice on the transmitter side without occurrence of sense of discomfort as to the voice being heard on the receiver side even in the event that a present environment is out of the ideal telephone call environment (telephone call environment with white noises), resulting in improvements in online speech quality. In addition, as the prediction coefficient from the background noise removal device is used to permit generation of the gain coefficient for adjustment of the gain of a pitch code vector and the gain coefficient for adjustment of the gain of a noise code vector in units of frame time periods to thereby adjust the gains of code vectors being output from the adaptive code book and noise code book, it becomes possible to hear and catch the voice on the transmitter side without creation of any uncomfortable feelings as to the voice being heard on the

receiver side even in the event that a present environment is out of the ideal telephone call environment (telephone call environment with white noises), thus enabling the over-the-phone voice quality to improve accordingly.

[Brief Description of the Drawings]

[Fig. 1] A principal block diagram of the inventions as recited in claim 1, claim 2 and claim 5.

[Fig. 2] A principle block diagram of the inventions as recited in claim 3 and claim 5.

[Fig. 3] A principle block diagram of the inventions as recited in claim 4 and claim 5.

[Fig. 4] A diagram showing one embodiment of the inventions as recited in claim 1, claim 2 and claim 5.

[Fig. 5] A diagram showing an embodiment of the inventions as defined in claim 3 and claim 5.

[Fig. 6] A diagram showing an embodiment of the inventions as set forth in claim 4 and claim 5.

[Fig. 7] A diagram showing one typical prior art A-b-S method-based high-efficiency audio/voice encoding apparatus.

[Explanation of Reference Numerals]

2 Noise Code Book

4 Selector Means

6 Analysis Means

8 Background Noise Removal Device

10 Inverse Filter

12	Power Integrator
14	Subtractor
16	Linear Prediction Coding Analysis Unit
18	Coefficient Calculation Unit
20	Pitch Gain Adjuster
21	Noise Gain Adjuster
22	Coefficient Difference Comparison Unit
23	Selector
30	Inverse Filer
34	Power Integrator
36	Subtractor
40	Inverse Filter
42	Power Integrator
44	Coefficient Calculation Unit
76_1	Noise Code Book
76_2	Noise Code Book
76_3	Noise Code Book
76_n	Noise Code Book

[Document Name] Drawings

[FIG. 1]

PRINCIPLE BLOCK DIAGRAM OF THE INVENTIONS AS RECITED IN CLAIMS 1-2 AND 5

ANALYSIS MEANS

NOISE CODE BOOKS SELECTOR MEANS

[FIG. 2]

PRINCIPLE BLOCK DIAGRAM OF THE INVENTIONS AS RECITED IN CLAIMS 3 AND 5

BACKGROUND NOISE REMOVAL DEVICE

INVERSE FILTER SUBTRACTOR

POWER INTEGRATOR

[FIG. 3]

PRINCIPLE BLOCK DIAGRAM OF INVENTIONS AS RECITED IN CLAIMS 4-5

8: BACKGROUND NOISE REMOVAL DEVICE

10: INVERSE FILTER 12: POWER INTEGRATOR

18: COEFFICIENT CALCULATOR 20: PITCH GAIN ADJUSTER

21: NOISE GAIN ADJUSTER

16: LINEAR PREDICTION CODING ANALYZER

[FIG. 4]

DIAGRAM SHOWING ONE EMBODIMENT OF THE INVENTIONS AS RECITED IN CLAIMS 1-2 AND 5

LPC COEFFICIENT   COEFFICIENT DIFFERENCE COMPARISON

NOISE CODE BOOKS   SELECTOR   NOISE CODE BOOK

[FIG. 5]

DIAGRAM SHOWING ONE EMBODIMENT OF THE INVENTIONS AS RECITED IN CLAIMS 3 AND 5

PREDICTION COEFFICIENT

INVERSE FILTER

POWER INTEGRATOR   NOISE CODE BOOK

[FIG. 6]

DIAGRAM SHOWING ONE EMBODIMENT OF THE INVENTIONS AS RECITED IN CLAIMS 4-5

PREDICTION COEFFICIENT   INVERSE FILTER   POWER INTEGRATOR

POWER VALUE COEFFICIENT CALCULATOR

[FIG. 7]

DIAGRAM SHOWING ARRANGEMENT OF PRIOR ART A-b-S METHOD-BASED HIGH-EFFICIENCY VOICE ENCODING APPARATUS

INPUT VOICE

ADAPTIVELY UPDATED

74: ADAPTIVE CODE BOOK   GAIN

70: LPC ANALYZER   72: SYNTHETIC FILTER   VOICE SYNTHESIZED

[FIXED]   76: NOISE CODE BOOK   96: ERROR MINIMIZATION